

User Manual

Four-Channel GSM VoIP Gateway

Model: GoIP4



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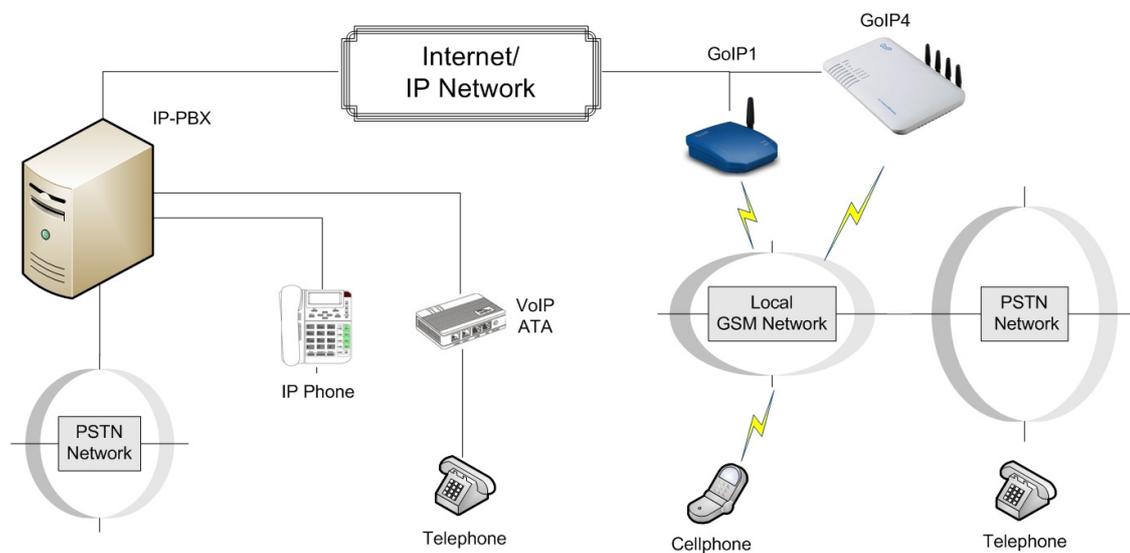
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1 Product Introduction

1.1 General Information

A VoIP GSM Gateway enables direct routing between IP and GSM network without the use of a FXO port or the PSTN network. With this device, the usage of VoIP is greatly enhanced with significant savings on long distance and roaming charges.



1.2 Protocol

- TCP/IP V4 (IP V6 auto adapt)
- ITU-T H.323 V4 Standard
- H.2250 V4 Standard
- H.245 V7 Standard
- H.235 Standard (MD5, HMAC-SHA1)
- ITU-T G.711 alaw/ulaw, G.729A, G.729AB, and G.723.1 Voice Codec
- RFC1889 Real Time Data Transmission
- Proprietary Firewall-Pass-Through Technology
- SIP V2.0 Standard
- Simple Traversal of UDP over NAT (STUN)
- Web-base Management
- PPP over Ethernet (PPPoE)
- PPP Authentication Protocol (PAP)

- Internet Control Message Protocol (ICMP)
- TFTP Client
- Hyper Text Transfer Protocol (HTTP)
- Dynamic Host Configuration Protocol (DHCP)
- Domain Name System (DNS)
- User account authentication using MD5
- Out-band DTMF Relay: RFC 2833 and SIP Info

1.3 Hardware Specification

- ARM9E Processor
- DSP for voice codec and voice processing
- Two 10/100 BaseT Ethernet ports with full compliant with IEEE 802.3
- LEDs for Ethernet port status
- Four GSM Channels' Connection

1.4 Software Specification

- LINUX OS
- Built-in HTTP Web Server
- PPPoE Dial-up
- NAT Broadband Router Functions
- DHCP Client
- DHCP Server
- Firmware On-line upgrade
- PSTN Caller ID transmit
- Multiple Language Support
- Supported call divert
- Supported PSTN auto call out to PSTN
- Supported Multi_devices Cooperate Mode(Group Mode)
- Supported SMS call out

1.5 List of the Package

- a) One GoIP4 Gateway main unit
- b) One DC12V/2000mA power adaptor
- c) One Ethernet cable (3M)

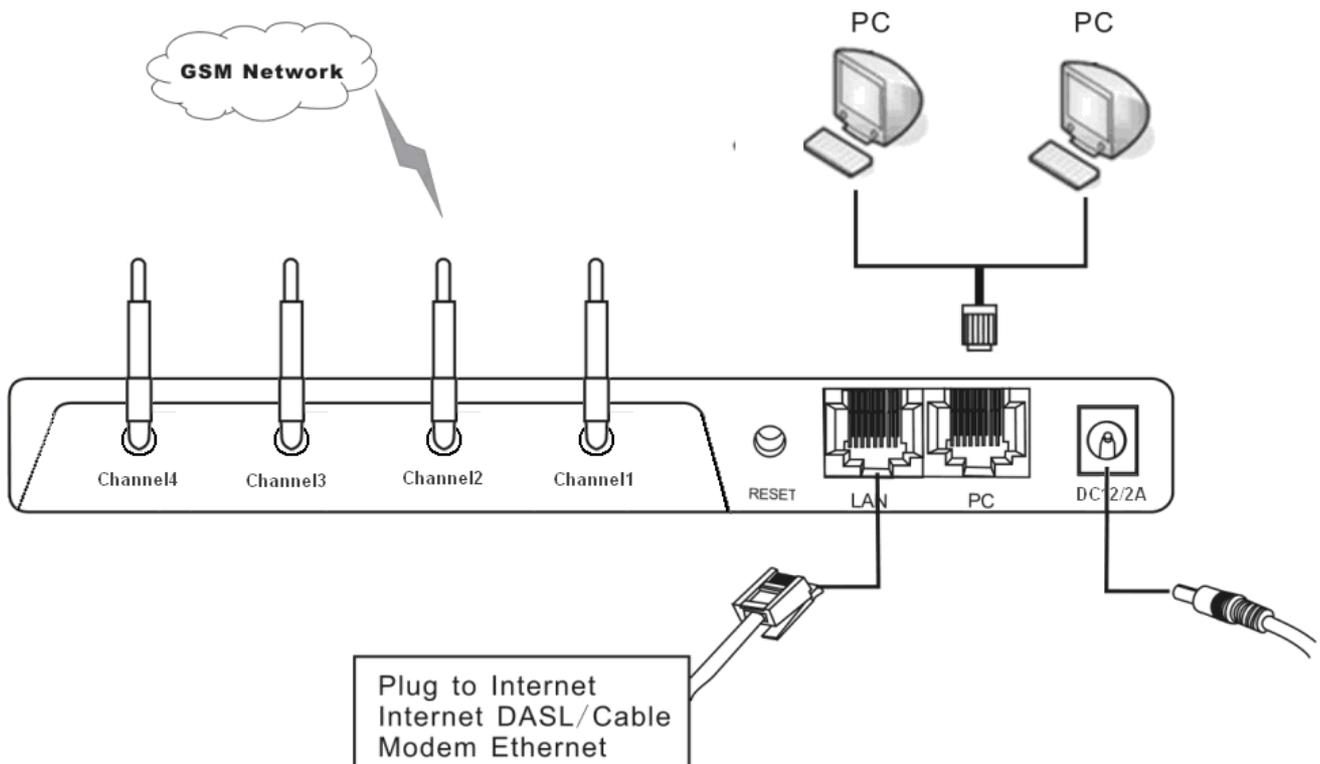
1.6 Appearance



- 1) **LAN**
Connect this port to an Ethernet Switch/Router, the Ethernet of a DSL modem, or other network access equipment.
- 2) **PC**
Connect a computer or other network device to this port.
- 3) **POWER (DC12V/2000mA)**
Connect the 12V/2000mA Adapter provided to this power jack.
- 4) **Reset**
Press this button to reset the GoIP_4 Gateway to factory defaults.

2 Installation

2.1 Installation Steps



Please follow the connection diagram above to install the GoIP4 Gateway.

- a) Insert a GSM SIM card in the SIM card compartment located at the bottom of the GoIP4 Gateway.
- b) Connect an Ethernet cable the LAN port of the GoIP4 Gateway and the other end to your existing network equipment.
- c) (Optional) Connect an Ethernet cable to the PC Port of the GoIP4 Gateway and the other end to a PC or other network device.
- d) Connect the power adapter provided to the power jack of the GoIP4 Gateway.

2.2 LED Indicators

The following table defines the status of the LEDs located on the top case and on the RJ-45 connectors.

LED	DESCRIPTION
POWER	When GoIP4 Working, this LED will keep light.
RUN	<ol style="list-style-type: none"> 1. When the GoIP4 is booting, this LED will flash 100ms ON and 100ms OFF. 2. When the GoIP4 is login your softswitch, this LED will flash 1s ON and 1s OFF.
LAN	When LAN port is connected to network, this LED will light and flash when has data through this port.
PC	When a device is connected to PC port, this LED will light and flash.
Channel 1	When the GoIP4's GSM channel1 login to ISP's system, this LED will flash 1s ON and 1s OFF;
Channel 2	When the GoIP4's GSM channel2 login to ISP's system, this LED will flash 1s ON and 1s OFF.
Channel 3	When the GoIP4's GSM channel3 login to ISP's system, this LED will flash 1s ON and 1s OFF.
Channel 4	When the GoIP4's GSM channel4 login to ISP's system, this LED will flash 1s ON and 1s OFF.

2.3 SMS Commands

GoIP4 supports some maintenance commands from SMS.

FUNCTION	SMS CONTENT	REMARK
Obtain LAN Port Info	INFO	Not case-sensitive
Reset device	RESET Password	Not case-sensitive



Four-Channel GSM VoIP Gateway

Reboot device	REBOOT Password	Not case-sensitive
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Note: In command **Reset** and **Reboot**, the Password is the GoIP4 device’s admin password. The command keywords can be majuscule and lowercase, but the password is case-sensitive.

- 1» Obtain LAN Port IP Address
Once the GSM SMS with message content “info” or “INFO” is received, the GoIP1 sends back a SMS message to the sender with the message content containing the LAN Port IP address.
- 2» Reset GoIP Configuration
Upon receiving the SMS message “RESET <password>” or “reset <password>”, the GoIP1 resets its configurations to factory defaults.
- 3» Reboot GoIP
Upon receiving the SMS message “REBOOT <password>” or “reboot <password>”, the GoIP1 reboots itself automatically.

Note: <password> is the webpage login password as described in Section 3.1.

3 Configuration Guide

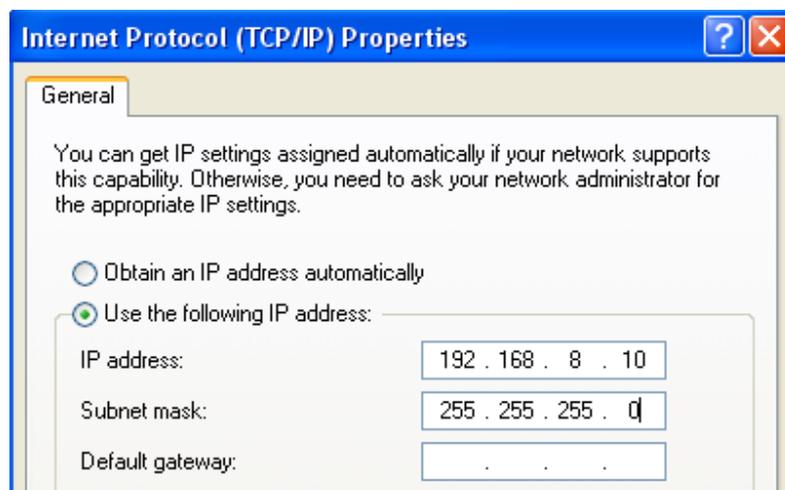
To configure the GoIP4 Gateway, you must login to its Web server via the LAN or PC port. The LAN port is factory preset to IP address 192.168.0.100 and the PC port is set to the fixed IP 192.168.8.1.

If you lose the IP address information for LAN port, just dial a call to GoIP4 Gateway’s SIM card phone number. When the call is connected, you will hear a dial tone. Then dial “*01#” for English voice prompt on the LAN IP and “*00#” for Chinese voice prompt on the LAN IP. The LAN IP Address can also be obtained by sending a SMS message to the GSM phone number. The GoIP4 will then reply with a SMS message containing the LAN IP address.

If you want to obtain LAN port IP by sending a SMS message, please send” INFO “or” info”.

Another way to access the GoIP4 Gateway is via the PC port. You will need to change your computer’s LAN configuration via the Network Connections under the Control Panel.

Windows: Control Panel-->Network Connections-->Local Area Connection Property--> Internet Protocol (TCP/IP)’s Property



Set an unused IP address that is in the same segment as the PC port address.

Once either the IP address of the LAN or PC port is known, you are now ready to access the Web server of GoIP4 Gateway.

3.1 Web Configuration Menu

If your computer is connected to the GoIP4 Gateway via the LAN port, you need to type the LAN IP address of the GoIP4 Gateway in your Web Browser to access the Web server of the GoIP4 Gateway. The default IP address on the LAN port is “192.168.0.100”.



If your computer is connected to the GoIP4 Gateway via the PC port, you should type GoIP4’s PC port IP address (192.168.8.1) in the Web Browser.



If the connection is correct, the Web Browser will prompt you to enter the “User name” and “Password” as shown below. Enter the User name and Password and the press OK to access the GoIP4 Gateway Web Server. The default for both user name and password is “admin”.



3.2 Status

The Status page shown below is the default / home page of the GoIP4 Web server.



Status						
Status Configurations Tools	Phone Information		Network Information		GSM Module Information	
	Serial Number	GOIP409100012	LAN Port	192.168.0.100	GSM1 Model	SIMCOM_SIM300
	Firmware Version	GHS-4.01-8-t3	LAN MAC	00:11:BE:02:F6:E0	GSM1 SIM	NOT INSERTED
	Hardware Model	GoIPx4	PC Port	192.168.8.1	GSM1 Signal 12	
	Line Register 1 Status	LOGOUT	PPPoE	Disabled	GSM1 Status LOGOUT	
	Line 1 State	IDLE	Default Route	192.168.0.1	GSM1 SIM Remain Time	NO LIMIT
	Line Register 2 Status	LOGOUT	DNS Server		GSM1 Number	
	Line 2 State	IDLE			GSM2 Model	SIMCOM_SIM300
	Line Register 3 Status	LOGOUT			GSM2 SIM	NOT INSERTED
	Line 3 State	IDLE			GSM2 Signal 12	
	Line Register 4 Status	LOGOUT			GSM2 Status LOGOUT	
	Line 4 State	IDLE			GSM2 SIM Remain Time	NO LIMIT
					GSM2 Number	
					GSM3 Model	SIMCOM_SIM300
					GSM3 SIM	NOT INSERTED
					GSM3 Signal 9	
				GSM3 Status LOGOUT		
				GSM3 SIM Remain Time	NO LIMIT	
				GSM3 Number		
				GSM4 Model	SIMCOM_SIM300	

3.2.1 Phone Information

A. Serial Number

Each Gateway has a unique serial number assigned by the factory such as **GOIP409100019**. This number is important for centralized configuration, technical support, and warranty. This number is printed on the bottom of the Gateway and is associated with your software license.

B. Firmware Version

Firmware version identifies the firmware version of the Gateway such as **GHS-4.01-1**.

C. Hardware Mode

This field shows terminal's hardware type.

D. Phone Status

This field shows the status of Line's connection status. If the connection is successful,

this field displays LOGIN; otherwise, it displays LOGOUT.

3.2.2 Network Information

A. LAN Port Configuration

This field displays the status of the LAN port.

B. PC Port Configuration

This field displays the status of the LAN port.

C. PPPoE

If PPPoE is enabled, it displays its status.

D. Default Route

This field displays the IP address of the default routing Gateway.

E. DNS Server

This field displays the IP address of the Domain Name Server.

3.2.3 GSM Module Information

A. GSM Module

This field displays the GSM module type.

B. GSM Signal

This field displays the GSM signal status. The value of GSM signal strength RSSI (Received Signal Strength Indication) is between 0 dbm and 31 dbm. The value of 99 means unknown or undetectable.

C. GSM Status

This field shows the status of GSM connection status. If the connection is successful, this field displays LOGIN; otherwise, it displays LOGOUT.

3.3 Configurations

Click on the “Configurations” tab on the left hand column to access the device configuration menu: **Preference, Network, Call Settings, Call Divert, Save Changes, and Discard Changes.**

Status

Configurations

- Preferences
- Network
- Call Settings
- Call Divert
- Save Changes
- Discard Changes

Tools

Preference

Time Zone	<input type="text" value="GMT+8"/>	Network Tones	<input type="text" value="United States"/>
Time Server	<input type="text" value="pool.ntp.org"/>	GSM CallerID Anonymous	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
DTMF Min Detect Time Gap	<input type="text" value="80"/>	GSM Band	<input type="text" value="850/1900"/>
Remote Control>>		<input type="checkbox"/> Auto Reboot	

Network Configuration

LAN Port	<input type="text" value="Static IP"/>	PC Port	<input type="text" value="Static IP"/>
IP Address	<input type="text" value="192.168.0.100"/>	IP Address	<input type="text" value="192.168.8.1"/>
Subnet Mask (optional)	<input type="text" value="255.255.255.0"/>	Subnet Mask	<input type="text" value="255.255.255.0"/>
Default Route	<input type="text" value="192.168.0.1"/>	DHCP Server	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Primary DNS	<input type="text"/>	Starting Address	<input type="text" value="192.168.8.100"/>
Secondary DNS (optional)	<input type="text"/>	Ending Address	<input type="text" value="192.168.8.120"/>
802.1q VLAN	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	Static DNS(optional)	<input type="text"/>
Advanced>>		Advanced>>	

Call Settings

Config Mode	<input type="text" value="Single Server Mode"/>	Advanced Settings>>
Phone Number	<input type="text"/>	Media Settings>>
Display Name	<input type="text"/>	
SIP Proxy	<input type="text"/>	
SIP Registrar Server	<input type="text"/>	

Click on “Preference” in the left menu of the configuration web, and the screen will be displayed as below:

Preference

Time Zone	<input type="text" value="GMT+8"/>	Network Tones	<input type="text" value="United States"/>
Time Server	<input type="text" value="pool.ntp.org"/>	GSM CallerID Anonymous	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
DTMF Min Detect Time Gap	<input type="text" value="80"/>	GSM Band	<input type="text" value="850/1900"/>
Remote Control>>		<input type="checkbox"/> Auto Reboot	

3.3.1 Language

Currently GoIP4 only supports English. VADcore also has other versions of software that support Simplified Chinese and Traditional Chinese. Contact VADcore if you need other language support.

3.3.2 Time Zone and Time Server

The GoIP4 Gateway supports Network Time Protocol (NTP) to obtain the date and time information from an NTP server (Time Server). The time zone is specified as in GMT ± offset. For example, the Pacific Standard Time is GMT-8, and the Pacific Daylight Time is GMT-7.

Time Zone	<input type="text" value="GMT+8"/>
Time Server	<input type="text" value="pool.ntp.org"/>

Note: The GoIP4 Gateway supports CDR and Billing Information, it is important to set up these two parameters properly.

3.3.3 DTMF Min Detect Time Gap

DTMF Min Detect Time Gap	<input type="text" value="50"/>
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This parameter is used to limit two same DTMF digit's minimum time gap, the range is 60ms to 120ms, default is 80ms.

If you encounter double digit problem, increase this parameter. If you encounter lose digit, then decrease this parameter.

3.3.4 Network Tone

Network Tones are a set of tones used for VoIP calls. Select one of the predefined countries or select "Customized" to define your own Network Tones.

Network Tones	<div style="border: 1px solid black; padding: 2px;"> <div style="border-bottom: 1px solid black; padding: 2px;">China ▼</div> <div style="padding: 2px;">Australia</div> <div style="padding: 2px;">China</div> <div style="padding: 2px; background-color: #e0e0e0;">Hong Kong</div> <div style="padding: 2px;">New Zealand</div> <div style="padding: 2px;">United Kingdom</div> <div style="padding: 2px;">United States</div> <div style="padding: 2px;">Customized</div> </div>
---------------	---

You can configure the Network Tones as Customized option:

Network Tones	<input type="text" value="Customized"/>
Dial Tone	<input type="text"/>
Ring Back Tone	<input type="text"/>
Busy Tone	<input type="text"/>
Indication Tone	<input type="text"/>

Each tone listed above is defined in the following format:

nc, rpt, c1on, c1off, c2on, c2off, c3on, c3off, f1, f2, f3, f4, p1, p2, p3, p4

Where:

- nc is the number of cadences
- rpt is the repeat counter(0 - infinite, 1-n - repeat 1-n times)
- c1on is the cadence one on (in milliseconds)
- c1off is the cadence one off (in milliseconds)
- c2on is the cadence two on (in milliseconds)
- c2off is the cadence two off (in milliseconds)
- c3on is the cadence three on (in milliseconds)
- c3off is the cadence three off (in milliseconds)
- f1 is the tone #1 frequency (300Hz-3000Hz)
- f2 is the tone #2, frequency (300Hz-3000Hz)
- f3 is the tone #3 frequency (300Hz-3000Hz)
- f4 is the tone #4 (300Hz-3000Hz)
- p1 is the attenuation index for f1, 0-31(0=3dB, -1dB increments)
- p2 is the attenuation index for f2, 0-31(0=3dB, -1dB increments)
- p3 is the attenuation index for f3, 0-31(0=3dB, -1dB increments)
- p4 is the attenuation index for f4, 0-31(0=3dB, -1dB increments)

For example, the tone definition for a tone of 450Hz with a cadence of 700 ms on and 1000 ms off is **1,0,700,1000,0,0,0,0,450,0,0,0,20,0,0,0**

3.3.5 GSM Caller ID Anonymous

GSM CallerID Anonymous Enable Disable

Some GSM ISPs allow the caller to disable the phone number (caller ID) when making outgoing calls. This feature must be supported by GSM ISPs.

3.3.6 GSM Band

GSM Band	<div style="border: 1px solid #ccc; padding: 2px;"> <div style="display: flex; justify-content: space-between; align-items: center;"> 900/1800 ▼ </div> <div style="background-color: #e6f2ff; padding: 2px;">900/1800</div> <div style="padding: 2px;">850/1900</div> </div>
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GolP4 Supported quad GSM bands: 850MHz, 900MHz, 1800MHz, 1900MHz. Select the correct GSM bands that are used in your country.

3.3.7 GSM ISP

GolP4 will show you a list of GSM ISPs When GolP4 completes a search of ISPs. You can choose the ISP from the list and GolP4 will just register to the one that you choose.

3.3.8 Auto Reboot

	<input checked="" type="checkbox"/> Auto Reboot
Reboot Time	<input type="text" value="04:00"/>

This option enables GolP4 to auto-reboot each day. GolP4 will auto-reboot itself at Reboot Time. If there is an active call, the action will be automatically delayed until the call is finished.

3.4 Call Settings

Click on the “Call Settings” to configure the VoIP call settings.

3.4.1 SIP Standard Supported

GolP4 supports SIP standard. GolP4 has three types of config modes for SIP protocol;

Call Settings	
Config Mode	<div style="border: 1px solid #ccc; padding: 2px;"> <div style="display: flex; justify-content: space-between; align-items: center;"> Single Server Mode ▼ </div> <div style="background-color: #e6f2ff; padding: 2px;">Multiple Server Mode</div> <div style="padding: 2px;">Trunk Gateway Mode</div> </div>
Phone Number	Single Server Mode
Display Name	Trunk Gateway Mode

Single Server Mode: All four channels use same SIP account to connect to SIP server.

Multiple Server Mode: Each channel use its own SIP account to connect to SIP server.

Trunk Gateway Mode: GolP4 will act as a SIP proxy. Remote SIP clients can register to GolP4 and GolP4 will process SIP requests on behalf of SIP client.

GolP4 Gateway’s SIP configure page as follow:

Call Settings	
Config Mode	Single Server Mode <input type="button" value="v"/>
Phone Number	<input type="text"/>
Display Name	<input type="text"/>
SIP Proxy	<input type="text"/>
SIP Registrar Server	<input type="text"/>
Register Expiry(s)	<input type="text"/>
Outbound Proxy	<input type="text"/>
Home Domain	<input type="text"/>
Authentication ID	<input type="text"/>
Password	<input type="text"/>
Call Forward Type	Not Forward <input type="button" value="v"/>
Call Forward Number	<input type="text"/>
Backup Server	<input type="radio"/> Enable <input checked="" type="radio"/> Disable

A) Phone Number

Enter a SIP phone number.

B) Display Name

Enter this field for the name to be displayed on the called VoIP party.

C) SIP Proxy

Enter the SIP proxy IP address or domain name. If the registration port is not 5060, then add “:” and the port number. For example: **sip.hybertone.com:8080**.

D) SIP Registrar Server

Enter the SIP registrar server IP address or domain name in this field. If the registration port isn’t 5060, add “:” and the port number. For example: **sip.hybertone.com:8080**.

E) Register Expiry(s)

Enter the register time (seconds) in this field. This is the maximum length of registration that SIP server will keep your registration. If SIP server does not receive another SIP

registration, the current registration will time out. Check your SIP server for a reasonable value.

F) Outbound Proxy

Outbound proxy is a device that receives requests from a client, even though it may not be the server resolved by the Request-URI. Outbound proxy will forward SIP requests and frequently RTP media traffic to another SIP server. Outbound proxy is used for a number of reasons, including, firewall traversal - both in parallel with a firewall and situated in the Internet as a Session Border Controller; and also for hiding customer IP addresses - calls are all routed through one point so that a public ITSP IP address can be used for accessing customers, rather than the customer's own IP address.

Check with your SIP server (SIP provider) if an outbound proxy is required.

G) Home Domain

SIP Networks sometimes use the Home Domain name as an identifier. Enter this field if it is required.

H) Authentication ID

Enter the Authentication ID as provided.

I) Password

Enter the authentication password as provided.

J) Call Forward Type

Call forward can be set under different conditions: Unconditional Forward, Busy Forward, No Answer Forward, Busy or No Answer Forward. Select the call forward type and enter the phone number that you would like the call to be forwarded to.

K) Call Forward Number

Enter the phone number that you would like the call to be forwarded to when Call Forward is set.

L) Backup Server

Backup Server	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Backup SIP Proxy	<input type="text"/>
Backup SIP Registrar	<input type="text"/>
Backup Home Domain	<input type="text"/>
Fail-retry Interval(1-60s)	<input type="text"/>

The GoIP4 Gateway supports one Backup Server as an alternative to the main server. When the registration to the main server fails, the GoIP4 Gateway will try to register to the Backup Server.

3.4.2 Advanced Settings

Click on “Advance Settings” tab on the top right corner of the Call Setting page to display all the parameters for programming, as shown below. These parameters allow more advanced control over the SIP signaling and media preference.

Advanced Settings<<

Local Signaling Port

SIP 183

NAT Keep-alive Enable Disable

Reigster Mode ▼

Advanced Timing>>

DTMF Signaling ▼

Outband DTMF type ▼

RTP Payload Type

Signaling QoS ▼

Signaling Encryption ▼

Signaling NAT Traversal ▼

Media Settings>>

A) Local Signaling Port (SIP Local port)

The default SIP port is 5060. Change this as required.

B) SIP 183

Check the box of SIP 183 if the SIP server supports this feature.

C) NAT Keep-alive

NAT Keep-alive Enable Disable

The NAT Keep-alive feature sends a null packet to the SIP Proxy periodically in order to keep the NAT open on your firewall for incoming data traffic.

3.4.3 Advanced Timing

Advanced Timing<<

No Answer Expiry(32-180s)	<input type="text" value="180"/>
NICT Expiry(2-180s)	<input type="text" value="2"/>
ICT Expiry(5-360s)	<input type="text" value="5"/>
Retransmit T1(200-2000ms)	<input type="text" value="200"/>
Retransmit T2(2000-8000ms)	<input type="text" value="2000"/>
DTMF Signaling	<input type="text" value="Outband"/>
Outband DTMF type	<input type="text" value="RFC 2833"/>
RTP Payload Type	<input type="text" value="101"/>
Signaling QoS	<input type="text" value="None"/>
Signaling Encryption	<input type="text" value="None"/>
Signaling NAT Traversal	<input type="text" value="None"/>

A) No Answer Expiry(32-180s), NICT Expiry(2-180s), Retransmit T1(200-2000ms), Retransmit T2(2000-8000ms)

Some SIP proxies may have special timing requirements. Change these parameters as required.

B) DTMF Signaling

1) DTMF TYPE

DTMF signals can be sent over to the called party after a call is established. GoIP4 Gateway supports both **Inband** and **Outband** DTMF signal types.

DTMF Signaling

Outband
Inband
Outband

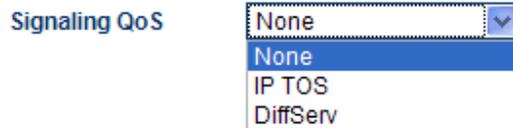
For **Inband** DTMF type, DTMF signals are generated locally at the calling phone and then send to the called party as part of the voice signals. This method is not reliable since the quality of the DTMF signals is subject to the Codec used and the quality of the network.

For **Outband** DTMF type, DTMF signals are independently translated and sent to the called party. After receiving DTMF signals, the called party translates and interpret based on the DTMF protocol. This method allows more reliable DTMF signaling. However, it requires the called party to support this feature in order for this to work properly. GoIP4 Gateway supports both RFC2833 and SIP INFO DTMF protocols.

2) DTMF Payload Type

DTMF Payload Type is defined by RFC2833 protocol to carry the tone definitions for various applications. The default DTMF Payload Type is 101. Please consult your VoIP service provider for the proper setting if required.

C) Signaling Qos



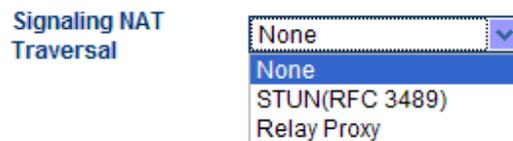
Signaling QoS improves the performance of SIP signaling. If local network device supports QoS, select this field accordingly. Please consult your network administrator for further information.

D) Signaling Encryption

GolP4 Gateway supports different encryptions for SIP signaling. Select the one that you prefer.

E) Signaling NAT Traversal

Signaling NAT traversal may be required if the GolP4 Gateway is put behind a NAT/firewall (or multiple NATs/firewalls). Depending on your network environment and SIP Server capabilities, this feature may or may not be turned on.



1) None

Select **None** to turn off this feature.

2) STUN (RFC 3489)

STUN (Simple Traversal of UDP (User Datagram Protocol) through NATs (Network Address Translators)) is a network protocol allowing a client behind a NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet-side port associated by the NAT with a particular local port.

Select STUN (RFC 3489) to use a STUN server for Signaling NAT Traversal. Enter the IP Address or the domain name of the STUN server to be used.

2) Relay Proxy

Relay proxy is a proprietary NAT traversal technology. Please consult your service provider for more information.

3.4.4 Media Setting

Click on “**Media Settings**” in the “**Call Setting**” menu to access the parameters available for media settings.

Media Settings<<<

RTP Port Range 16384 - 32768

PacketLength(ms) 20

Jitter Buffer Fixed ▾

Delay(ms) 60

Media QoS None ▾

Media Encryption None ▾

Symmetric RTP

Media NAT Traversal None ▾

A) RTP Port Range

This parameter specifies the range of the RTP (Real Time Protocol) Ports used by the GoIP4 Gateway. If your network limits the usable port range, this parameter may need to be modified. Please consult your network administrator for more information.

B) Packet Length(ms)

This parameter defines the voice packet length. The default setting is 20ms. The range is from 5ms to 40ms at an increment of 5ms. Please note that some codes have a minimum packet length of more than 5ms.

C) Jitter Buffer Mode

Jitter Buffer Mode Fixed ▾

Minimum Jitter

Maximum Jitter(soft limit)

Since data packets may arrive at different orders, the Jitter Buffer is used to hold the data packets received for re-arrangement according to the packet sequence number. Three Jitter Buffer Modes are supported: Adaptive, Sequential, and Fixed. The default is set to Fixed mode with the fixed delay of 60ms. Please consult your network administrator for more information on the network environment in order to determine the optimal settings.

D) Media Qos

Media QoS None ▾

None
IP TOS
DiffServ

Similar to the Signaling QoS, the Media QoS is intended to improve the voice performance or quality if your local network supports QoS.

E) Media Encryption

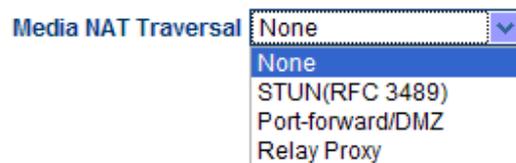
GolP4 Gateway supports different encryptions for voice media. Select the one that you prefer.

F) Symmetric RTP

Normally GolP4 Gateway uses RTP ports based on the configuration. If this box is checked, GolP4 Gateway will identify RTP ports from the media traffic it has received and use the same ports when sending media traffic.

G) Media NAT Traversal

Similar to Signaling NAT Traversal, this feature allows media packets (RTP) to be routed properly in various network environments.



1) None

Select **None** to disable this feature.

2) STUN (RFC 3489)

STUN (Simple Traversal of UDP (User Datagram Protocol) through NATs (Network Address Translators)) is a network protocol allowing a client behind a NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet-side port associated by the NAT with a particular local port.

Select STUN(RFC 3489) to use a STUN server for Signaling NAT Traversal. Enter the IP Address or the domain name of the STUN server to be used.

3) Port forwarding Support

Port forwarding (sometimes referred to as tunneling) is the act of forwarding a network port from one network node to another. This technique can allow an external user to reach a port on a private IP address (inside a LAN) from the outside via a NAT-enabled router.

In order for this feature to work, the local network Gateway must support this feature and be set up properly. Please consult your network administrator for help to enable this Port forwarding feature.

4) Relay Proxy

Relay proxy is a proprietary NAT traversal technology. Please consult your service provider for more information.

Currently, the following 3 kinds of packaging mechanism are supported:

- Mode 1: The media uses UDP packets and (or) encrypt with multiple UDP port;
- Mode 2: The media uses UDP packets and (or) encrypt with single UDP port;
- Mode 3: The media uses TCP packets and (or) encrypt (UDP over TCP).

Media NAT Traversal

Address

Port

User Name

Password

Encryption

Relay Mode

Backup Relay Server 1

Backup Relay Server 2

Backup Relay Server 3

Backup Relay Server 4

3.4.5 Codec Preference

Click on “Media Settings” in the “Call Setting” menu and click **Audio Codec Preference** to access the parameters.

Codec Preference allows a user to select the codes to be used and its priority for a voice call.

Audio Codec Preference<<

UP

DOWN

- alaw
- ulaw
- g729
- g729a
- g729ab
- g7231

Click on the check box to enable a codec. Select a codec and then press the UP or DOWN button to move the position of the codec on the codec list with a priority in descending order.

Note: The voice code alaw and ulaw is G.711a and G.711u.

3.5 Call Divert

The Call divert feature controls the routing of calls between VoIP and GSM.

Call Divert

CID Forward Mode Use CID as SIP Caller ▾

Line1
 Line2
 Line3
 Line4

Forward to PSTN
 Enable
 Disable

Forward Number (VoIP To PSTN)

Dial Plan (VoIP to PSTN)

Forward to PSTN Auth Mode No Auth ▾

SMS Sender

SIM Card Settings>>

VoIP Trust List>>

Forward to VoIP
 Enable
 Disable

Forward Number (PSTN To VoIP)

Dial Plan (PSTN to VoIP)

Forward to VoIP Auth Mode No Auth ▾

PSTN Trust List>>

GSM Group Mode Disable ▾

SMS Mode Disable ▾

3.5.1 Call Forward (From VoIP To PSTN)

Line1
 Line2
 Line3
 Line4

Forward to PSTN
 Enable
 Disable

Forward Number (VoIP To PSTN)

Dial Plan (VoIP to PSTN)

Forward to PSTN Auth Mode No Auth ▾

SMS Sender

SIM Card Settings>>

VoIP Trust List>>

Forward Number

Enter a phone number in this field will forward all incoming VoIP calls to this phone number (PSTN or Mobile). Using “,” to add a 500ms delay to the dialing sequence.

When Forward Number field has a phone number, GoIP4 will automatically forward all VoIP calls to this phone number.

When Forward Number is empty, GoIP4 will route phone calls based on the following conditions.

A: When the Callee ID is GoIP4’s SIP account number, GoIP4 will take the call and feed back a dial tone to VoIP caller. Then VoIP caller must dial a PSTN number when hearing this dial tone.

B: When the Callee ID is not GoIP4’s SIP account number, GoIP4 will automatically dial out with this number thru GSM network, based on the rules in Dial Plan(VoIP to PSTN) field.

Dial Plan

Please refer to **3.9 Dial Plan** for details. If “:” is entered in the field, all of the phone calls will pass through.

Forward to PSTN Auth Mode

This field sets protection for using GoIP4 to connect to GSM network.

- 1) No Auth
Anyone can make phone calls through GoIP4.
- 2) Password
If a password is entered, the GoIP4 will generate an indication tone and wait for the caller to dial the password.
- 3) Trust List

Forward to PSTN Auth Mode Trust List

SMS Sender

[SIM Card Settings>>](#)

[VoIP Trust List<<](#)

VoIP Trust List

Trust Number1	<input type="text"/>
Trust Number2	<input type="text"/>
Trust Number3	<input type="text"/>
Trust Number4	<input type="text"/>
Trust Number5	<input type="text"/>
Trust Number6	<input type="text"/>

Enter the phone numbers on the Trust Number field if Trust List is used. People calling from the trust phone numbers will be able to use GSM connection.

4) Password or Trust List

Callers will be able to use GoIP4 for GSM connection if their phone numbers are on trust phone number list or if they have the password.

SIM Card Settings

SIM Card Expiry	<input type="text"/>
SIM Card State Report Number	<input type="text"/>
SIM Card State Report Time	<input type="text" value="30"/>
SIM Card ID	<input type="text"/>

- 1) SIM Card Expiry - usage limit (minutes)
- 2) SIM Card State Report Number - the recipient phone number for the SMS report
- 3) SIM Card State Report Time - the time schedule to send SMS report
- 4) SIM Card ID - Identification sent with the sms message

3.5.2 Call Forward (From PSTN To VoIP)

Forward to VoIP	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Forward Number (PSTN To VoIP)	<input type="text"/>
Dial Plan(PSTN to VoIP)	<input type="text" value=":"/>
Forward to VoIP Auth Mode	<input type="text" value="No Auth"/> ▾
	PSTN Trust List>>
GSM Group Mode	<input type="text" value="Disable"/> ▾
SMS Mode	<input type="text" value="Disable"/> ▾

Forward Number

Enter a phone number in this field will forward all incoming PSTN (GSM) calls to this phone number (a VoIP number).

If this field is blank, the GoIP4 answers all incoming GSM calls and then generates the dial tone. The caller can then dial a VoIP number. When finishing, a pound (#) can be dialed to activate the dialing to the VoIP number immediately. If a pound (#) is not input, the VoIP number will be dialed after a preset timeout.

When

Dial Plan

Please refer to **3.9 Dial Plan** for details. If “:” is entered in the field, all of the phone calls will pass through.

Forward to VoIP Auth Mode

This field sets protection for using the GSM connection to VoIP.

- 5) No Auth
Anyone can make phone calls through GoIP4.
- 6) Password
If a password is entered, the GoIP4 will generate an indication tone and wait for the caller to dial the password.
- 7) Trust List

Forward to VoIP Auth Mode ▼

PSTN Trust List<<

PSTN Trust List

Trust Number1

Trust Number2

Trust Number3

Trust Number4

Trust Number5

Trust Number6

Enter the phone numbers in the Trust Number field if Trust List is used. People calling from the trust phone numbers will be able to use GoIP4 to connect to VoIP.

- 8) Password or Trust List
Callers will be able to use GoIP4 for VoIP connection if their phone numbers are on trust phone number list or if they have the password.

3.6 SMS Disposal

3.6.1 SMS Sender

SMS Sender

SMS Server IP

SMS Server Port

SMS Client ID

Password

VADcore offers a software to send out SMS to GSM network through GoIP4 Gateway. A SMS

server is required to work with GoIP4 Gateway for SMS Sender. Please contact VADcore for more details.

3.6.2 SMS Call Out

GoIP4 Gateway supported SMS call. In this mode, when GoIP4 Gateway receives a SMS from a mobile phone, it will automatically make a call to SIP server.

To use this function, select the SMS Dial option in configuration page.

PSTN Trust List>>

GSM Group Mode	<input type="text" value="Disable"/>
SMS Mode	<input type="text" value="Dial"/>
SMS Dial	<input type="text" value="Mode 1"/>
SMS Dial Prefix	<input type="text"/>

GoIP4 supported three types SMS Dial:

GSM Group Mode	<input type="text" value="Disable"/>
SMS Mode	<input type="text" value="Dial"/>
SMS Dial	<input type="text" value="Mode 1"/>
SMS Dial Prefix	<input type="text" value="Mode 1"/> <input type="text" value="Mode 2"/> <input type="text" value="Mode 3"/>

A: Mode 1

GoIP4 dial the call use SMS sender call ID

B: Mode 2

GoIP4 dial the call via its VoIP account and add the SMS sender phone number to Call Divert option's Forward Number (VoIP to PSTN) automatically.

C: Mode 3

GoIP4 dial the call via its VoIP account and add the SMS sender phone number to SIP invites header.

D: SMS Dial Prefix

When GoIP4 dial a SMS call, it will automatically add this option's digit in be Called ID.

- **Mode 1 examples:**

A. GoIP4 use SMS Dial Mode 1:

PSTN Trust List>>

GSM Group Mode	<input type="text" value="Disable"/>
SMS Mode	<input type="text" value="Dial"/>
SMS Dial	<input type="text" value="Mode 1"/>
SMS Dial Prefix	<input type="text"/>

A mobile phone's number is (86)13800000000, it sends a SMS "8675588228822" to GoIP4's GSM SIM card. When GoIP4 device receives this SMS, it will automatically call the number 8675588228822, and the caller is number 8613800000000.

The sent-out signaling as follow:

```

Sending Message to 192.168.2.1:5060:␣
INVITE sip: 8675588228822@192.168.2.1:5060;transport=udp SIP/2.0␣
Via: SIP/2.0/UDP 192.168.2.189:5060;rport;branch=z9hG4bK1686911003␣
From: <sip: 861380000000@192.168.2.1:5060>;user=phone;tag=626918067␣
To: <sip: 8675588228822@192.168.2.1>␣
Call-ID: 1835068843@192.168.2.189:5060␣
CSeq: 2 INVITE␣
Contact: <sip: 861380000000@192.168.2.189:5060>␣
Max-Forwards: 30␣
User-Agent: HyberTone␣
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER,
MESSAGE, INFO, SUBSCRIBE␣
Content-Type: application/sdp␣
Content-Length: 226␣
    
```

B. GoIP4 use SMS Dial Mode 1 and add a prefix as 999:

SMS Mode	<input type="text" value="Dial"/>
SMS Dial	<input type="text" value="Mode 1"/>
SMS Dial Prefix	<input type="text" value="999"/>

A mobile phone's number is (86)13800000000, it sends a SMS "8675588228822" to GoIP4's GSM SIM card. When GoIP4 device receives this SMS, it will automatically call the number 9998675588228822, and the caller is number 8613800000000.

The sent-out signaling as follow:

```

Sending Message to 192.168.2.1:5060:␣
INVITE sip: 9998675588228822@192.168.2.1:5060;transport=udp SIP/2.0␣
Via: SIP/2.0/UDP 192.168.2.189:5060;rport;branch=z9hG4bK1686911003␣
From: <sip: 861380000000@192.168.2.1:5060>;user=phone;tag=626918067␣
To: <sip: 9998675588228822@192.168.2.1>␣
Call-ID: 1835068843@192.168.2.189:5060␣
CSeq: 2 INVITE␣
Contact: <sip: 861380000000@192.168.2.189:5060>␣
Max-Forwards: 30␣
User-Agent: HyberTone␣
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER,
MESSAGE, INFO, SUBSCRIBE␣
Content-Type: application/sdp␣
Content-Length: 226␣

```

- Mode 2 example:

GoIP4 use SMS Dial Mode 2:

PSTN Trust List>>

GSM Group Mode	<input type="text" value="Disable"/>
SMS Mode	<input type="text" value="Dial"/>
SMS Dial	<input type="text" value="Mode 2"/>
SMS Dial Prefix	<input type="text"/>

A mobile phone's number is (86)13800000000, it sends a SMS "8675588228822" to GoIP4's GSM SIM card. When GoIP4 device receives this SMS, it will automatically call the number 8675588228822, and the caller number is GoIP4's SIP account number.

GoIP4 will set the SMS sender number to "Call Divert "option's "Forward Number (VoIP to PSTN)" automatically. The result is, when SIP server receives the SMS call and call back to GoIP4, GoIP4 will automatically call the SMS sender via GSM network.

The sent-out signaling as follow:

Sending Message to 192.168.2.1:5060:␣
 INVITE sip: 8675588228822@192.168.2.1:5060;transport=udp SIP/2.0␣
 Via: SIP/2.0/UDP 192.168.2.189:5060;rport;branch=z9hG4bK92531725␣
 From: <sip:20001@192.168.2.1:5060>;user=phone;tag=740569827␣
 To: <sip: 8675588228822@192.168.2.1>␣
 Call-ID: 464713443@192.168.2.189:5060␣
 CSeq: 3 INVITE␣
 Contact: <sip:20001@192.168.2.189:5060>␣
 Max-Forwards: 30␣
 User-Agent: HyberTone␣
 Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER,
 MESSAGE, INFO, SUBSCRIBE␣
 Content-Type: application/sdp␣
 Content-Length: 226␣

SMS prefix can be used in mode 2 just like in mode 1.

- Mode 3 example:

GoIP4 use SMS Dial Mode 3:

PSTN Trust List>>

GSM Group Mode	<input type="text" value="Disable"/>
SMS Mode	<input type="text" value="Dial"/>
SMS Dial	<input type="text" value="Mode 3"/>
SMS Dial Prefix	<input type="text"/>

A mobile phone's number is (86)13800000000, it sends a SMS "8675588228822" to GoIP4's GSM SIM card. When GoIP4 device receives this SMS, it will automatically call the number 8675588228822*(86)13800000000, and the caller number is GoIP4's SIP account number.

The sent-out signaling as follow:

```

Sending Message to 192.168.2.1:5060:↵
INVITE sip: 8675588228822*861380000000@192.168.2.1:5060;transport=udp
SIP/2.0↵
Via: SIP/2.0/UDP 192.168.2.180:5060;branch=z9hG4bK620642232↵
From: <sip:20001@192.168.2.1:5060>;user=phone;tag=1333994780↵
To: <sip: 8675588228822*861380000000@192.168.2.1>↵
Call-ID: 52754291@192.168.2.180↵
CSeq: 2 INVITE↵
Contact: <sip:20001@192.168.2.180:5060>↵
Max-Forwards: 30↵
User-Agent: HyberTone↵
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER,
MESSAGE, INFO, SUBSCRIBE↵
Content-Type: application/sdp↵
Content-Length: 226↵

```

SMS prefix can be used in mode 3 just like in mode 1.

3.6.3 SMS Relay

GoIP4 GSM Gateway supports SMS relay.

PSTN Trust List>>

GSM Group Mode	<input type="text" value="Disable"/>
SMS Mode	<input type="text" value="Relay"/>
SMS Forward Number	<input type="text"/>

The **SMS Forward Number** is the receiver (ex. an extension) on your VoIP system. GoIP4 will forward the SMS it received to the number.

3.6.3.1 SMS Relay To VoIP System

When GoIP4 receives a SMS from GSM network, it will relay to VoIP system’s appointed number (SMS Forward Number).

Assume the SMS Forward Number is 3999 and SMS sender number is “8613682626865 “, the SMS content is “075583185700 “. The GoIP4 will send a message to your VoIP system as below:

```
MESSAGE sip:3999@192.168.2.1 SIP/2.0↵
Via: SIP/2.0/UDP 192.168.2.162:5060;branch=z9hG4bK1967685528↵
From: <sip:1638@192.168.2.1>;tag=667435795↵
To: <sip:3999@192.168.2.1>↵
Call-ID: 2094144847@192.168.2.162↵
CSeq: 4 MESSAGE↵
Contact: <sip:1638@192.168.2.162:5060>↵
Max-Forwards: 30↵
User-Agent: HyberTone↵
Content-Type: text/plain↵
Content-Length: 28↵
↵
8613682626865↵
075583185700↵
```

3.6.3.2 SMS Relay To GSM Network

When GolP4 receives a message from SIP server as below:

```
MESSAGE sip:1638@192.168.2.162:5060 SIP/2.0↵
From: <sip:3999@192.168.2.89>;tag=5031↵
To: <sip:1638@192.168.2.1>↵
Call-ID: 808807EB-A8B3-DD11-BBA6-005056C00008@192.168.2.89↵
CSeq: 3 MESSAGE↵
Contact: <sip:3999@192.168.2.89>↵
max-forwards: 16↵
date: Tue, 18 Nov 2008 06:36:37 GMT↵
user-agent: SIPPER for 3CX Phone↵
p-hint: usrloc applied↵
Content-Type: text/plain↵
Content-Length: 26↵
↵
13682626800↵
Hello world↵
```

The GolP4 will send a SMS to GSM number **13682626800**, the SMS content is “**Hello world**”.

3.7 GSM Caller ID Transparent

GolP4 supports GSM Caller ID transparent to VoIP via SIP Invite signaling.

CID Forward Mode	Use CID as SIP Caller <input type="button" value="v"/>
	Disable
	Use Remote Party ID
	Use CID as SIP Caller ID

- A) **Disable:** Disable GSM Caller ID transparent to VoIP.
- B) **Use Remote Party ID:** GolP4 will add Caller ID in SIP invite's Remote Party ID option.

```

Sending Message to 192.168.2.1:5060:
INVITE sip:5000@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.180:5060;branch=z9hG4bK1645487913
From: <sip:20001@192.168.2.1:5060>;user=phone;tag=406202416
To: <sip:5000@192.168.2.1>
Call-ID: 847230278@192.168.2.180
CSeq: 2 INVITE
Contact: <sip:2000@192.168.2.180:5060>
Max-Forwards: 30
User-Agent: HBT
Remote-Party-ID: "13800000000"
<sip:13800000000@192.168.2.1>;party=calling;screen=no;privacy=off
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER,
MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length: 226

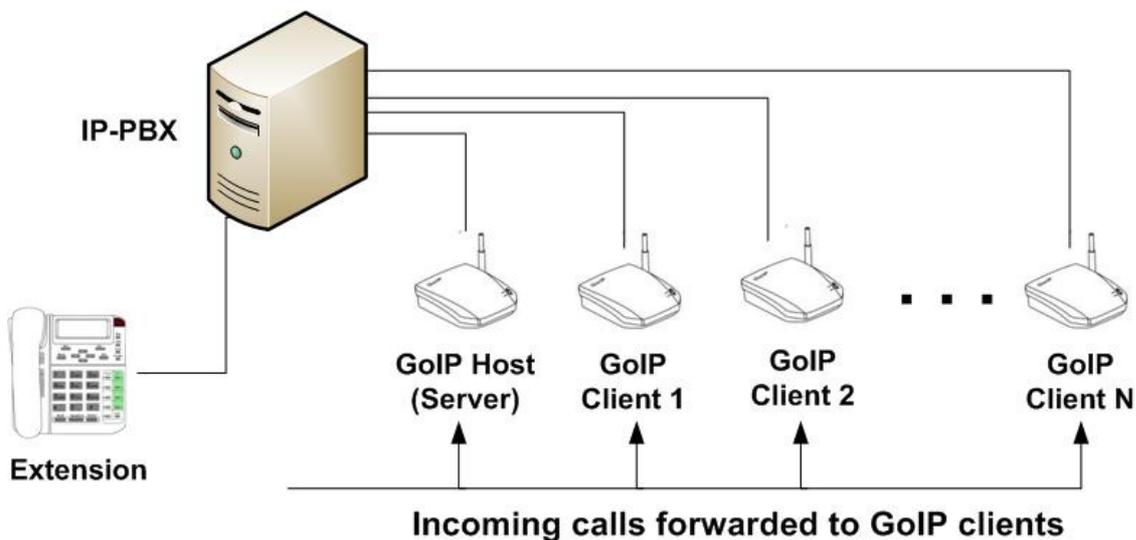
```

- C) **Use CID as SIP Caller ID:** GolP4 use PSTN Caller ID in SIP invitee's Caller ID option and Remote Party ID option.

```

Sending Message to 192.168.2.1:5060:
INVITE sip:5000@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.180:5060;branch=z9hG4bK1450498491
From: "13800000000" <sip:13800000000@192.168.2.1:5060>;tag=232569343
To: <sip:5000@192.168.2.1>
Call-ID: 1853068986@192.168.2.180
CSeq: 2 INVITE
Contact: <sip:13800000000@192.168.2.180:5060>
Max-Forwards: 30
User-Agent: HBT
Remote-Party-ID: "13800000000" <sip:
13800000000@192.168.2.1>;party=calling;screen=no;privacy=off
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER,
MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length: 226
    
```

3.8 GSM Group Mode



GoIP4 can group multiple devices together and provide line-hunt. The Group Mode works like a multi-channels GSM gateway. Any GoIP4's channel can work as Group Server Mode or Client Mode.

GSM Group Mode	As Server ▼
SMS Mode	Disable As Server As Client

Server Mode:

Only one GSM channel runs in **Server Mode**. The GSM channel that is set in Server Mode will forward the GSM's incoming calls to other available client channels. The GSM channel that is set in Server Mode will be your main number for your customer.

Client Mode:

Other GSM channels will run in **Client Mode**. The GSM channels that are set in Client Mode will report their status to GSM channel that is set in Server Mode. The GSM channel in Server Mode then forwards phone calls to available GSM channels in Client Mode.

You must enter the GSM number for that GSM channel and IP address of the device in Server Mode into the field.

GSM Group Mode	As Client ▼
Server Address	
GSM Number	

Disable: Please set all channels to **Disable Mode** if you would like to run each channel independently.

3.9 Dial Plan

Dial Plan defines how a number is processed when GoIP4 receives it. This field is located in the **Call Divert** Window. The Dial Plan is very flexible and can be configured for a wide range of dialing applications.

Call Divert

Line1
 Line2
 Line3
 Line4

Forward to PSTN
 Enable
 Disable

Forward Number (VoIP To PSTN)

Dial Plan(VoIP to PSTN)

Forward to PSTN Auth Mode

SMS Sender
[SIM Card Settings>>](#)
[VoIP Trust List>>](#)

Forward to VoIP
 Enable
 Disable

Forward Number (PSTN To VoIP)

Dial Plan(PSTN to VoIP)

Forward to VoIP Auth Mode

[PSTN Trust List>>](#)

The basic syntax is “<event>:<action>|<event>:<action>|...”, where

<event> defines the event to be matched. An event consists of a sequence of digits. If a specific digit has a limited range, use the syntax [A-B] where A and B are both digit (0 to 9) and B is greater than A. The length of the input number can be limited by using “X” to represent each unknown digit. If this field is omitted, it means any event.

<action> defines the action to be taken when a phone number is received. It consists of “-“ (minus), “+” (plus), and digits. “-“ followed by digits means to remove the digits from the beginning of the number. “+” followed by digits means to add the digits in front of the number.

“|” means “or” and the order of priority is from left to right.

Note: For practical use, there should be no limitation on the length of dial plan string.

Examples:

1. Dial Plan = “010:-010” means that the first 3 digits ”010” of dialed number will be removed if the first 3 digits of dialed number are “010”..
 - a) Number entered = “01082121234”, actual number dialed = “82121234”.
 - b) Number entered = “82121234”, actual number dialed = “82121234”.

2. Dial Plan = "1:+00" means that two digits "00" will be added in front of the number when the first digit of the dialed number is "1".
 - a) Number entered = "1082121234", actual number dialed = "00182121234".
 - b) Number entered = "82121234", actual number dialed = "82121234".

3. Dial Plan = "001:-001+1751" means that the first 3 digits "001" of the dialed number will be changed to "1751" when a number with first three digits "001" is entered.
 - a) Number entered = "00182121234", actual number dialed = "175282121234".
 - b) Number entered = "82121234", actual number dialed = "82121234".

4. Dial Plan = "XXXX:" means that the input number is limited to 4-digit long and will be dialed out immediately when the fourth digit is entered.

5. Dial Plan = "13XXXXXXXXX:+0" means that the input number is restricted to 11-digit long and the first two digits must be "13". When this condition is matched, the digit "0" will be added to the front of the number and then dialed out.
 - a) Number entered = "13901234567", actual number dialed = "013901234567".
 - b) Number entered = "12801234567", actual number dialed = "12801234567".

6. Dial Plan = "13[6-9]XXXXXXXXX:+0" means that the input number is restricted to 11-digit long, the first two digits must be "13" and the third digit can be 6, 7, 8, or 9. When this condition is matched, the digit "0" will be added to the front of the number and then dialed out.
 - a) Number entered = "13901234567", actual number dialed = "013971234567".
 - b) Number entered = "13001234567", actual number dialed = "13001234567".

Please note that the above samples are intended to show the meaning of various rules. They may not have any practical meaning. A combination of these rules (joined with the symbol "|") can be realized for a much more complicated dialing application.

3.10 Gain Settings

A hidden webpage is provided to set the receiving and transmitting gains of VoIP Channel. The URL link is:

http://xxx.xxx.xxx.xxx/vadcore/en_US/gain.html

THIS PAGE IS INTENDED FOR AN EXPERIENCED USER OR AN ADMINISTRATOR ONLY. PLEASE SET THE GAINS WITH CAUTIONS.

Note: A too low or too high gain MAY affect the sensitivity of DTMF detections.



3.11 Network Configuration

Click on "Network" tab in the left menu column to configure the LAN and PC ports.

Network Configuration			
LAN Port	Static IP	PC Port	Static IP
IP Address	192.168.0.100	IP Address	192.168.8.1
Subnet Mask (optional)	255.255.255.0	Subnet Mask	255.255.255.0
Default Route	192.168.0.1	DHCP Server	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Primary DNS		Starting Address	192.168.8.100
Secondary DNS (optional)		Ending Address	192.168.8.120
802.1q VLAN	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	Static DNS(optional)	
Advanced>>		Advanced>>	

3.11.1 LAN Port

Three LAN Port modes are supported: **DHCP**, **Static IP** and **PPPoE**. The default is set for Static IP with default IP address “192.168.0.100”.

Network Configuration	
LAN Port	Static IP
IP Address	DHCP
Subnet Mask (optional)	Static IP
Default Route	PPPoE
Primary DNS	255.255.255.0
Secondary DNS (optional)	
802.1q VLAN	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Advanced<<	
Ethernet(MAC) Address	
IP Broadcast Address	

1) DHCP

Choose **DHCP** if a local DHCP host is available. This allows the GoIP4 Gateway to obtain network information (IP Address, Subnet Mask, Default Route, Primary DNS, Secondary DNS, and other DHCP options) from the DHCP host.

2) Static IP

Network Configuration	
LAN Port	Static IP <input type="button" value="v"/>
IP Address	<input type="text" value="192.168.0.100"/>
Subnet Mask (optional)	<input type="text" value="255.255.255.0"/>
Default Route	<input type="text" value="192.168.0.1"/>
Primary DNS	<input type="text"/>
Secondary DNS (optional)	<input type="text"/>
802.1q VLAN	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Advanced<<	
Ethernet(MAC) Address	<input type="text"/>
IP Broadcast Address	<input type="text"/>

The default setting of GoIP4 is **Static IP** with IP address “192.168.0.100” and **Subnet Mask** “255.255.255.0”. However, **Default Route**, **Primary DNS**, and **Secondary DNS** (optional) must be manually entered according to your network configuration.

3) PPPoE

Network Configuration	
LAN Port	PPPoE <input type="button" value="v"/>
User Name	<input type="text"/>
Password	<input type="text"/>
802.1q VLAN	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Advanced<<	
Ethernet(MAC) Address	<input type="text"/>
IP Broadcast Address	<input type="text"/>

PPPoE is a common method for you network modem (Cable / xDSLs). Choose this if your network environment requires. Enter the **User Name** and **Password** as provided by your ISP.

4) 802.1q VLAN

This QoS feature requires QoS support of your network to improve voice traffic. Please consult your network administrator for proper settings.

5) Advanced...

The **Advanced** settings allow the user to set the broadcast address and to clone a MAC address instead of using the factory preset MAC address. Please consult your network administrator for further information.

3.11.2 PC Port Configurations

The PC Port allows other network devices to be attached to the GoIP4 Gateway. It offers both Bridge and Static IP modes to meet your requirements. The factory preset is Static IP mode with the IP address 192.168.8.1.

PC Port	Static IP
IP Address	192.168.8.1
Subnet Mask	255.255.255.0
DHCP Server	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Starting Address	192.168.8.100
Ending Address	192.168.8.120
Static DNS(optional)	
	Advanced<<
Ethernet(MAC) Address	
IP Broadcast Address	

1) Bridge Mode

Select **Bridge** mode if your network topology requires the network devices (PC or others) to be in the same network segment as the GoIP4 Gateway. In this case, the GoIP4 Gateway functions as an Ethernet Switch.

2) Static IP Mode (Default Setting)

Select **Static IP** mode for a new network segment on PC port. In this case, the GoIP4 Gateway functions as Router. Enter the IP address in **IP Address** field with a new segment address that is different from that on the LAN port. Enter the subnet mask in **Subnet Mask** field accordingly. A commonly used value is 255.255.255.0.

PC Port	Static IP
IP Address	192.168.8.1
Subnet Mask	255.255.255.0

Enable the **DHCP Server** if you want the GoIP4 Gateway functions as a local DHCP host on PC port. This will enables the GoIP4 Gateway to assign IP Addresses to network devices that are attached to the PC port.

DHCP Server	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Starting Address	<input type="text" value="192.168.8.100"/>
Ending Address	<input type="text" value="192.168.8.120"/>
Static DNS(optional)	<input type="text"/>

Specify the **Starting Address**, **Ending Address**, and **Static DNS** accordingly.

4) **Advanced**

The **Advanced** settings allow the user to set the broadcast address and to clone a MAC address instead of using the factory preset MAC address. Please consult your network administrator for further information.

Advanced<<

Ethernet(MAC) Address	<input type="text"/>
IP Broadcast Address	<input type="text"/>

3.12 Save Configuration

To confirm and commit all changes that have been made, click on the **Save Changes** tab. Otherwise, all changes will be discarded.

3.13 Discard Changes

To discard all changes made, click on the **Discard Changes** tab.

3.14 Tools Menu

Select the **Tools** to access the following functions: **Online Upgrade**, **Change Password**, **Reset Config**, and **Reboot**.

<p>Status</p> <p>Configurations</p> <p>Tools</p> <p>Online Upgrade</p> <p>Change Password</p> <p>Reset Config</p> <p>Reboot</p>	<p>Online Upgrade</p>
	<p>Last Upgrade Time:</p> <p>Current Version: GHS-4.01-8-t3</p> <p>Upgrade Site: <input type="text"/> <input type="button" value="Start"/></p>

3.14.1 Online Upgrade

To perform a firmware upgrade, select the **Online Upgrade** tab to access the page below.

<p>Online Upgrade</p>
<p>Last Upgrade Time:</p> <p>Current Version: GHS-4.01-8-t3</p> <p>Upgrade Site: <input type="text"/> <input type="button" value="Start"/></p>

Enter the update link as provided by VADcore. A sample link is:

hk.ippcn.com/update/GHS-3.01-18.pkg

Click the **Start** button to start the firmware upgrade.

WARNING: POWER SHUTDOWN, POWER FAILURE OR UNPLUG POWER ADAPTOR FROM GoIP4 DURING FIRMWARE UPGRADE MAY PERMINENTLY DAMAGE THE GOIP4 GATEWAY AND VOID THE WARRANTY.

3.14.2 Change Password

Click on the **Change Password** tab to access the page below.

User Level

New Password:

Confirm Password:

Administration Level

New Password:

Confirm Password:

A) User Password

This is the password for the user ID “user”. The default password is “1234”. This user ID has limited access to the Network Configuration menu.

B) Administrator Password (default: admin)

This is the password for the user ID “admin”. The default password is “admin”. This user ID has full access to all configuration settings available.

3.14.3 Reset Configuration

Click on the **Reset Config** tab to reset the GolP4 Gateway to its factory default settings.

3.14.4 Reboot the Device

Click on the **Reboot** tab to reboot the GolP4 Gateway.

4 Hardware Specifications

Characteristics of the hardware	Parameter	Remarks
Processor	ARM9E 133MHz	
DSP	VPDSP101-4 100MHz	
RAM	16M	
Flash	4M	
Power	DC12V/2000mA +-10%	Input AC100V to AC240V
GSM Module Type	Default 850MHz/1900MHz	
	Optional 900MHz/1800MHz	
Consumption	The Maximum 5 W	
LEDs	RUN, GSM, LAN, PC,GSM	
Network Ports	2 RJ45; Supported NAT	100/10BASE-T
Weight	900 Grams	Full Set
Working Temperature	0—40°C	
Working Humidity	40%—90% Not Congealed	
Colour	Grey	
GSM SIM Ports	4	
VoIP Channels	4	

5 Manufactory Parameters

Parameters		Default Setting
Network	LAN	192.168.0.100
	PC	Static IP:192.168.8.1 DHCP Server Running
Password	admin	admin
	user	1234



Four-Channel GSM VoIP Gateway

Time Zone	GMT+8